

HandyTone-701 User's Manual 1.1



Thank you for purchasing Grandstream HT-701. Before using the product, please read the manual carefully. After reading, store the manual in an accessible place.

Check if all the accessories/attachments are included in the package.

- HT-701
- AC Adaptor
- Ethernet cable
- User's manual

DISCLAIMER

- Features and specifications may be updated without prior notice. Please bear with us.
- The company will not be responsible for the development of noise and volume interruption and disconnection of calls that arise from the customer and this company's network system. Please be advised that call quality is not guaranteed.
- The company will not be responsible for inconsistency in call quality brought about by individual listening skills.
- The company will not be responsible for the device damage that occurred outside Japan.
- Please acknowledge that services in HandyTone-701 may stop temporarily without prior notice due to HT-701's scheduled maintenance.
- Please acknowledge that the company will not be responsible in the event that the phone can no longer be used for calling due to damage or malfunction.

ATTENTION

- Wipe the device with soft and dry cloth. Do not use wet towel.
- Do not use products with chemicals as these may cause damage to the device.
- If chemical dust cloth is used, read the accompanying directions carefully.

● *HandyTone-701*

HandyTone-701 is an adaptor used to connect an analog phone device to *Agile Phone*. To use the device, the customer needs to subscribe to an *Internet Service Provider*. Customer with no broadband router features needs to provide a separate broadband router.



Port	Description
INTERNET	Connects LAN cable's (Ethernet) accessory.
DC 12V	Connects special AC adaptor accessory.
PHONE	Connects analog phone device. ※ Use appropriate phone cable accessory to analog phone when connecting.
RESET	Use when restarting the device.

● *Connection Method*

1. Switch the analog phone device's line type to *tone*.
2. Connect the phone cable with HandyTone-701's *Phone* and analog phone device.
3. Connect Ethernet cable with HandyTone-701's *Internet* to a broadband router.
4. Connect AC Adaptor to HandyTone-701's *DC 12V* and plug to an electric socket.

※ Perform the procedure in correct order. Make sure that cable and adaptor were plugged in properly. It takes a couple of minutes to place a call after connection. Please wait a moment.

•Setting from Web Browser

① Confirm HandyTone-701 device's IP address.

1. After picking-up the handset of the phone device that is connected to HandyTone-701, dial ***.
2. If setting is not yet performed, E103E, an English voice, will be played from the handset. After dialing #, confirm whether an English voice saying *ENTER A MENU OPTION* is played.

※ If already set, a voice saying *ENTER A MENU OPTION* will be played.

3. After *ENTER A MENU OPTION* is played, dial 02. An English voice telling HandyTone-701's IP address will be played.

Example: 1-9-2.1-6-8.1.2

② From web browser, open Setting page.

1. From the HandyTone-701 and computers connected within the same network, when launching Internet Explorer, etc. web browser, IP address should be entered following the format below.

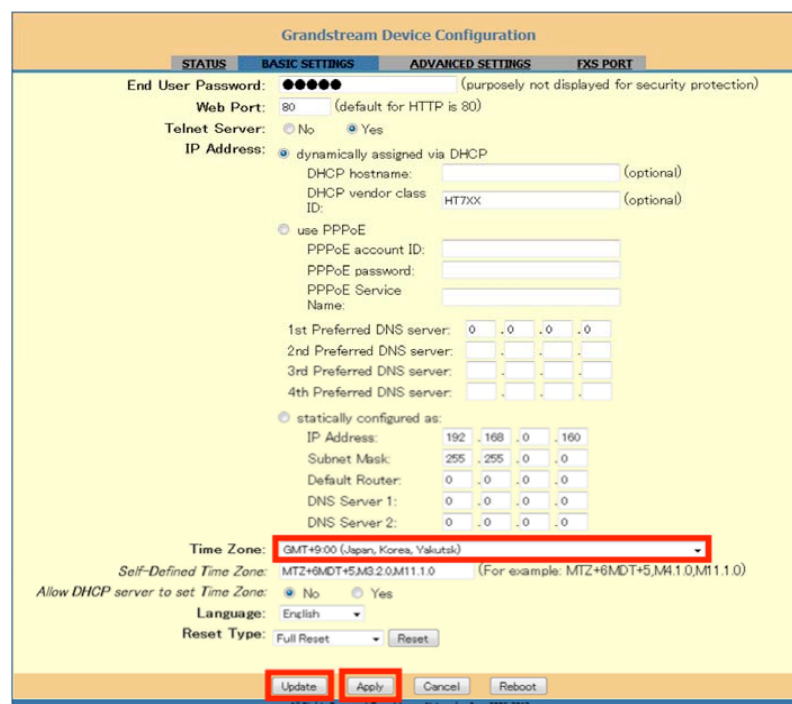
http://(IP Address)/

Example: If 192.168.1.2 is displayed, it should be entered as http://192.168.1.2/

2. The above page will be displayed. Enter password and click *Login*. The initial password is *admin*.



③ Click the *BASIC SETTINGS* tab found at the upper portion of the page.



Time Zone : Select *GMT+9:00 (Japan,Korea,Yakutsk)*.

Once done, click *Update* and *Apply* respectively.

- ④ Click the **ADVANCED SETTINGS** tab found at the upper portion of the page.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT

Account Active: ☐ No ☒ Yes

Primary SIP Server: **voip30xx.agile.ne.jp** (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Prefer Primary SIP Server: ☐ No ☒ Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy: **voip30xx.agile.ne.jp** (e.g., proxy.myprovider.com, or IP address, if any)

SIP Transport: ☒ UDP ☐ TCP ☐ TLS (default is UDP)

NAT Traversal (STUN): ☐ No ☒ No, but send keep-alive ☐ Yes

SIP User ID: **UID** (the user part of an SIP address)

Authenticate ID: **UID** (can be identical to or different from SIP User ID)

Authenticate Password: **端末パスワード** (purposefully not displayed for security protection)

Name: **UID** (optional, e.g., John Doe)

Incoming SIP Messages from SIP Proxy Only: ☒ No ☐ Yes

SIP T1 Timeout: 0.5 sec

SIP T2 Interval: 4 sec

DTMF Payload Type: 101

Preferred DTMF method (in listed order):
Priority 1: In-audio
Priority 2: RFC2833
Priority 3: In-audio

Disable DTMF Negotiation: ☒ No (negotiate with peer) ☐ Yes (use above DTMF order without negotiation)

Send DTMF as SIP Event: ☐ No ☒ Yes (Hook Flash will be sent as a DTMF event if set to Yes)

Distinctive Ring Tone: Ring Tone 1 (yes, call features using star codes will be supported locally)
Ring Tone 1 (used if offhook) (User ID/extension to dial automatically when offhook)
Ring Tone 1 (used if incoming)

Disable Call-Waiting: ☒ No ☐ Yes

Disable Call-Waiting Caller ID: ☐ No ☒ Yes

Disable Call-Waiting Tone: ☐ No ☒ Yes

Disable Receiver Offhook Tone: ☐ No ☒ Yes (ROH tone will not be played after offhook for 60 seconds)

Disable Reminder Ring for On-Hold Call: ☐ No ☒ Yes

Disable Visual MWI: ☐ No ☒ Yes

Ring Timeout: 60 (10-300, default is 60 seconds)

Delayed Call Forward Wait Time: 20 (Allowed range 1-120, in seconds)

No Key Entry Timeout: 4 (in seconds, default is 4 seconds)

Early Dial: ☒ No ☐ Yes (use "Yes" only if proxy supports 484 response)

Dial Plan Prefix: (this prefix string is added to each dialed number)

Use # as Dial Key: ☐ No ☒ Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)

Dial Plan: {x+|*x+|*xx*x+}

SUBSCRIBE for MWI: ☐ No, do not send SUBSCRIBE for Message Waiting Indication
☒ Yes, send periodical SUBSCRIBE for Message Waiting Indication

Send Announce: ☐ No ☒ Yes (caller ID will be blocked if set to Yes)

Force INVITE: ☐ No ☒ Yes

Send Re-INVITE After Fax: ☐ No ☒ Yes (90 seconds)

Enable 100rel: ☐ No ☒ Yes (90 seconds)

Use First Matching Vocoder in 200OK SDP: ☐ No ☒ Yes

Preferred Vocoder (in listed order):
choice 1: PCMU
choice 2: PCMA
choice 3: PCMU
choice 4: PCMU
choice 5: PCMU
choice 6: PCMU

G723 Rate: ☒ 6.3kbps encoding rate ☐ 5.3kbps encoding rate

DTMF After Fax Tone Detected: ☒ Enabled ☐ Disabled

Jitter Buffer Type: ☐ Fixed ☒ Adaptive

Jitter Buffer Length: ☐ Low ☒ Medium ☐ High

SRTP Mode: ☒ Disabled ☐ Enabled but not forced

SLIC Setting: USA 1 (BELLCORE 600 ohms)

Caller ID Scheme: **NTT Japan**

Polarity Reversal: ☒ No ☐ Yes (reverse polarity upon call establishment and termination)

Loop Current Disconnect: ☒ No ☐ Yes (loop current disconnect upon call termination)

Ring Tone 6: c=2000/4000; (range, minimum: 300 maximum: 1100)

Ring Tone 7: c=2000/4000; (range, minimum: 300 maximum: 1100)

Ring Tone 8: c=2000/4000; (range, minimum: 300 maximum: 1100)

Ring Tone 9: c=2000/4000; (range, minimum: 300 maximum: 1100)

Ring Tone 10: c=2000/4000; (range, minimum: 300 maximum: 1100)

Update Apply Cancel Reboot

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Primary SIP Server: Enter *VoIP server*.

Outbound Proxy: Enter *VoIP server*.

NAT Traversal (STUN): Select *No, but send keep-alive*.

SIP User ID: Enter *Unique* (10 numeric characters)

Authenticate ID: Enter *Unique* (10 numeric characters)

Authenticate Password: Enter *device password*.

Name: Enter *Unique* (10 numeric characters)

Preferred DTMF method: Supply *Priority 1, 2 and 3* with the same information as below:

Priority 1: In-audio

Priority 2: RFC2833

Priority 3: In-audio

Disable Call-Waiting :

- Selecting **Yes** will notify new caller that line is busy when the callee is still at the middle of the call.
- Selecting **No** will notify new caller to assume that the line is ringing/active even when the callee is at the middle of a call.

※ This is an optional setting so setting this is not necessary.

※ During delivery, this will become **No**.

Select Indication. Preferred Vocoder: Supply *Choice 1-8* with the same information as below:

Choice1: PCMU

Choice2: PCMA

Choice3: PCMA

Choice4: PCMA

Choice5: PCMU

Choice6: PCMA

Choice7: PCMA

Choice8: PCMA

Caller ID Scheme: Set using the example below.

- **NTT Japan** : Select when connecting to a phone device that supports number display

※ It is important to set phone that supports number display. Please refer to the device's instruction manual for the setup method.

- **Bellcore/Telcordia** : Select this when using phone other than *NTT Japan*.

※ The prescribed value is *Bellcore/Telcordia*. The default factory setting is *NTT Japan*. If the device does not support number display, change the setting to *Bellcore/Telcordia*.

Once done, click *Update* and *Apply* respectively.

- ⑤ Click the **ADVANCED SETTINGS** tab found at the upper portion of the page.

Grandstream Device Configuration

STATUS BASIC SETTINGS **ADVANCED SETTINGS** FXS PORT

Admin Password: (purposely not displayed for security protection)

Layer 3 QoS: 48 (Diff-Serv or Precedence value)

802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)

Confirmation value (NATed traffic) 0 (0-7)

Call Waiting Tone: (URI or IP:port)

Syntax: f1=val1[,f2=val2] (Frequencies are in Hz and cadence)

Lock Keypad Update: ☒ No ☐ Yes (configuration update via web is disabled if set to Yes)

Disable Voice Prompt: ☒ No ☐ Yes (voice prompt is disabled if set to Yes)

Disable Direct IP Call: ☒ No ☐ Yes (direct IP call is disabled if set to Yes)

NTP Server: **ntp.jst.mfeed.ad.jp** (URI or IP address)

Allow DHCP option 42 to override NTP server: ☒ No ☐ Yes

Syslog Server:

Syslog Level: NONE

Send SIP Log: ☒ No ☐ Yes

Download Device Configuration: Download

Upload firmware: Upload from local directory

Update Apply Cancel Reboot

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NTP Server: Enter *ntp.jst.mfeed.ad.jp*.

Once done, click *Update* and *Apply* respectively. Click *Reboot*. The device will be restarted and settings will be applied.

• **Important points when using Agile Network**

Always *disable the alert info header setting* in *Unique* that you are using on the device. Incoming call will be blocked when this setting is enabled.

From website, open *Unique Management* page.

Click *Unique Setting*, then, go to *General Setting*.

In *Alert Info Header Setting*, select *DISABLE*.

• Confirmation Setting from English Voice Guidance

1. Pick-up the handset.
2. From HandyTone-701 connected phone device, enter * * *. From the handset, a voice guidance saying *ENTER A MENU OPTION* will be played. When you dial *, confirming the status, *DHCP mode* or *STATIC IP mode*, is possible.
3. Enter menu number from phone device and perform confirmation of setting.
4. As needed, dial 9 and set the operation.

Menu Number	English Voice Guidance	Operating procedure/Contents
01	DHCP MODE (Dynamic Host Configuration Protocol Mode) and STATIC IP MODE	Warning: Static IP mode is not supported in Agile Phone service.
02	IP ADDRESS	Voice guidance for the current HandyTone's IP address will be played.
03	SUBNET	The current subnet mask's IP address will be played in voice guidance.
04	GATEWAY	Voice guidance for the current gateway's IP address will be played.
05	DNS SERVER	Voice guidance for the current DNS server's IP address will be played.
	INVALID ENTRY	Returns to Main Menu automatically.

Safety Precautions

This manual contains precautions to assure user's safety while using this product.

If the precautions are disregarded, the extent of consequences is presented below.











This indicates that ignoring or mishandling of this notice might result to death or serious injury to the person.









This indicates that ignoring or mishandling of this notice might result to harm to the person.



-  Immediately unplug the device from the power supply if there is a strange smell and smoke. Don't use the device and immediately contact the dealer for repair. This may cause fire or explosion.
-  Do not disassemble or modify. The device may malfunction, cause electric shock and fire.
-  Do not insert metal to the opening or drop the product. If in case a metal is inserted to the device, unplug the device and contact the dealer. The device may malfunction, cause electric shock and fire.
-  Do not pour water or any foreign object inside the device.
If in case water or any foreign object is poured or inserted to the device, unplug the device and contact the dealer.
-  Do not touch the device when lightning occurs. This may cause electric shock.
-  Do not use in bathroom or shower room. The device may malfunction, cause electric shock and fire.
-  Don't place in a damp, dusty or direct sunlight. The device may malfunction, cause electric shock and fire.
-  This device is not waterproof. If in case water entered the device, unplug the device and contact the dealer.



-  Do not place near the kitchen table or humidifier, or near oil and moisture. The device may malfunction, cause electric shock and fire.
-  Do not place near direct sunlight. The internal temperature might go up and may cause fire.
-  Do not place in humid and dusty area. The device may malfunction, cause electric shock and fire.
-  Do not place in extremely cold place or place with unstable temperature. It may cause explosion.
-  Do not place in unstable or shaky place.
If the device falls it may collapse, it may cause injury or explosion.
-  Use after checking if foreign matters are attached to the speaker of the handset. Thumbtacks, pin, stapler wires and other metallic object may cling to the magnet of the speaker so please be careful.

Troubleshooting

Please check the troubleshooting procedure before requesting a repair. If the problem persists, please contact Agile Shop at (sup@agile.ne.jp).

PROBLEM	CAUSE	SOLUTION
General Failure	<p>Please plug and unplug from the power source. (Do this a few times).</p> <p>Please wait at least one minute before you again plug the device to the power source.</p>	
No ringer sound	<p>Ethernet cable is disconnected.</p> <p>AC adapter is disconnected.</p> <p>Did you change the volume settings?</p> <p>Did you change the settings on the display?</p>	<p>Please check the connection of the Ethernet cable.</p> <p>Please check the connection of the AC adapter.</p> <p>Please adjust the volume on the volume button.</p> <p>Revert the changed settings.</p> <p>Disconnect the Ethernet cable (reboot).</p>
Call is interrupted by noise	Are there any abnormalities in the network environment?	Please check your network environment.

1. From web browser, open *Setting* page and *Login*.
2. Click *BASIC SETTINGS* tab found at the upper portion of the page.

The screenshot shows the 'Grandstream Device Configuration' web interface. The 'BASIC SETTINGS' tab is selected. The 'Reset Type' dropdown menu is set to 'Full Reset', and the 'Reset' button is highlighted with a red box. Other visible settings include 'End User Password' (masked with dots), 'Web Port' (80), 'Telnet Server' (No), 'IP Address' (dynamically assigned via DHCP), and 'Time Zone' (GMT+9:00 (Japan, Korea, Yakutsk)).

3. Reset Type: Immediately after selecting *Full Reset*, press the *Reset* button.
4. The phone device will be restarted and the phone will be initializing.

For more information on features and settings, please check <http://www.agile.ne.jp/support/>

Inquiries by telephone and email

Send to: sup@agile.ne.jp

Telephone number: 636- 5-150603

Weekdays: 10:00 - 19:00

